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TITLE OF THE INVENTION
VOICE OPTIMIZATION IN A NETWORK HAVING VOICE OVER INTERNET
PROTOCOL COMMUNICATION DEVICES

20   CROSS-REFERENCE TO RELATED PATENT APPLICATIONS
U.S. Serial No. 09/220,232, filed December 23, 1998 and entitled
WIRELESS MULTI-SITE NETWORKING USING SIGNALING AND VOICE OVER
INTERNET PROTOCOL, now pending.

25   BACKGROUND OF THE INVENTION

Field of the Invention
     The invention relates to optimizing voice quality on
networks that have voice over Internet Protocol communication
30   devices. The optimization is based on taking and evaluating
measurements of network performance pertaining to quality of the
network connection.

Discussion of Related Art

       The contents of U.S. Serial No. 09/220,232 is
incorporated herein by reference.

       Support for end-to-end voice calls using the Internet

5   as an alternative to traditional public switched telephone
networks (PSTN) is conventional. Unlike the PSTN, which is
circuit-switched, the Internet is packet-switched; communication
on the Internet is accomplished by transmitting and receiving
packets of data.

10   In addition to data, each packet contains an address to
ensure that it is routed correctly. The format of these packets
is defined by the Internet Protocol (IP). One type of allowable
data is encoded, digitized voice. Thus, voice over IP (VOIP) is
voice that is packetized as defined by IP, and communicated over

15   the Internet or some other IP network for telephone-like
communication.

       Devices such as personal computers (PC) that are VoIP
ready, wireless VoIP devices, intelligent phones that are VoIP
ready, conferencing units, trunks that are VoIP ready, multi-

20   media devices that are VoIP ready, and plain old telephone system
(POTS) telephones connected to VoIP gateways may be VoIP end-
points. Examples include Microsoft's NetMeeting, Nortel
Networks's soft-client and IP based personal digital assistant
(PDA) devices.

25   These end-points require real-time performance over IP
networks. The end points are assumed to be distributed on one or
more IP networks. Some devices will be on the same subnet on the
same local area network (LAN) segment while others will be on
different subnets, some of which could be at the end of a slow

30   wide area network (WAN) connection. A problem arises, however,
with respect to voice quality and other quality of service (QoS)
parameters in an IP network.

       Conventionally, bandwidth is reserved for particular IP
end points by marking their packets with a priority level.

However, about 90% of the networks today are not policy managed. With respect to the 10% that are policy managed, there are no solutions to the problems that exist when the bandwidth is insufficient to support a badly configured IP end-point.

5   Conventional software tools are available to help assess the performance of a network by measuring parameters of network performance. Such tools include a Ping tool, a Network trace tool and a packet loss measurement tool. These tools are standard with MICROSOFT WINDOWS product. They run from the

10 command line (found under Start, Run on all WINDOWS operating systems). The commands are 'ping IP_Address' or 'Trace IP_Address'. These tools are available with TCP/IP protocol stacks found under operating systems such as Linux/VxWorks, etc.

          With the telecommunications networks transitioning from

15 circuit switching to packet switching, phones, trunks and wireless devices are expected to be VoIP capable. Controlling these phones, trunks, and wireless devices to optimize media performance is desired.

20 BRIEF SUMMARY OF THE INVENTION

          An aspect of the invention resides in an apparatus and a method of optimizing voice quality on a network having end-point devices, such as voice over Internet Protocol (IP) devices. The invention includes initializing default parameters for the

25 end-point devices with respect to choice of preferred CODEC, number of voice samples per IP packet and jitter buffer size. Further, performance parameters are measured and if warranted by an indication from the performance parameters that the connection to the network is below a desired level of operation, the default

30 parameters are adjusted. The measurements include network bandwidth, number of network hops to the IP end-point devices, round trip delay and packet loss over a time duration. Software tools used to obtain the measurements may include a ping tool, a network trace tool and a packet loss measurement tool. The

adjustment of the default parameters may involve re-negotiating a CODEC connection, re-setting the packet size and/or re-setting the jitter buffer size.

5   Preferably, a private branch exchange is on the network. A register is configured to register the end-point VOIP devices with the private branch exchange (PBX) on the network. In response to the register completing registration of the end-point devices with the PBX, a terminal proxy server may set the default parameters of the registered end-point devices on the basis of

10   measured performance parameters. Thereafter, the default parameters are adjusted as necessary to better optimize the connection on the basis of changes to the performance parameters as measured by the software tools.

BRIEF DESCRIPTION OF THE SEVERAL VIEWS OF THE DRAWING

15   For a better understanding of the present invention, reference is made to the following description and accompanying drawing, while the scope of the invention is set forth in the appended claims.

          Fig. 1 shows a flow chart representing a condition of

20   the invention at initialization time.

          Fig. 2 shows a flow chart representing a three-phase approach in accordance with the invention.

          Fig. 3 shows a schematic representation of IP telephony in accordance with the invention.

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DETAILED DESCRIPTION OF THE INVENTION

          Before discussing the figures 1-3, a brief discussion about various terms used in this patent application may be helpful in better understanding the invention.

30   Bandwidth usually refers to maximum available bit rate for a specific application. In a specific embodiment, synchronous, interactive, and real-time applications, which are bandwidth-sensitive, can require minimum bandwidth guarantees, and can require sustained and burst-scale bit-rates. On the other

hand, network administrators may want to limit bandwidth taken by non-productive traffic. However, even though bandwidth may be allocated for specified applications, it does not mean that these applications may be using that bandwidth.

5   Jitter generally refers to variation in delay (e.g., the delay is not constant for all packets of a given flow) for a particular application. Real-time applications have maximum jitter limits. Applications such as real-audio and video do some advanced buffering in an attempt to overcome variation in packet

10   delays--the amount of buffering is determined by the expected jitter.

          A jitter buffer is a variable length buffer that stores voice packets. The size of the jitter buffer may be dependent upon the variation in the end-to-end packet delay in the network

15   and can be dynamically adjusted based upon, for example, this packet delay variation.

          Latency generally refers to the delay experienced by a packet from the source to destination. Latency requirements are typically specified as mean-delay and worst case delay in some

20   cases. Delay is a result of propagation delay, due to physical medium and queuing at intermediate nodes such as routers, gateways, servers, etc. A certain portion of the delay can be controlled by changing the way in which the queues are serviced at the intermediate nodes, and by controlling congestion at

25   bottleneck points. Some examples of delay measurements are packet round-trip delay and connection response time.

          A Ping tool includes a routine that sends a packet onto the network and obtains a value of the average delay encountered by that packet in reaching the destination and returning.

30   A network trace tool includes a routine that captures and records events and states that occur while the network is operating. Events in a trace chronology may be paired, forming event-pairs. Examples of event-pairs include the start and end times.

A packet loss measurement tool measures the rate of packet loss over a period of time. Packet loss is a rate of loss in a packet or a portion of packets that is generally caused by noise or failure of network elements (e.g., routers, servers) to

5   forward or deliver packets. Packet loss is usually an indication of severe congestion, overload of an element, or element failure (e.g., if a server is down). Even if the packet was not dropped but just delayed, protocols and applications may assume it was lost. Packet loss can cause application timeouts, loss of quality

10 or delay due to retransmitted packets. Indirect results of packet loss may also be measured (e.g., connection retries or data retransmits).

The default parameters of VoIP end-point devices need to be initialized (Fig. 1) and adjusted later as necessary (Fig.

15 2). Fig. 3 illustrates an example of VoIP end-point devices on a LAN. These devices include a wired VoIP Client (Set B), a wired soft VoIP Client (Set C) and Wireless VoIP CLIENTS (SETS D & E). These identified end-point devices are for illustration purposes only, there can be more or less end-point devices of different

20 types and additional LANs and additional subnets, etc.

The LAN is connected to a WAN via a call server, terminal proxy server and router in a conventional manner. A wired VoIP Client (Set A) is in communication with one or more of the VoIP devices of the LAN via the WAN. The parameters that may

25 be optimized include CODEC selection, packet size (that is, the number of frames per packet), desired latency, packet loss, available bandwidth, number of router hops, and jitter buffer size.

When a VoIP end-point device registers with an IP PBX,

30 the IP PBX may perform a number of tests to determine the optimum configuration for that end-point. It can measure network bandwidth, the number of network hops (if any) to the end point, the round trip delay and/or the packet loss over a short period of time. These tests can easily be done using a ping tool, a

network trace tool and/or a packet loss measurement tool. After
initialization, one or more of these tools can be used on regular
periods to determine network performance. During active
communications, a call re-establishment can be initiated

5   (manually or automatically) to re-apply these parameters.

Turning to Fig. 1, at the time of initialization, the
VoIP end-point device registers with the PBX. In response to the
registration, a terminal proxy server uses a protocol to instruct
the end-point device to use a CODEC of a particular type, a

10   jitter buffer of a particular size, a frame size of so many voice
samples, etc. That is, the terminal proxy server sets the
default settings for that VoIP end-point device. When the VoIP
end-point device attempts to make a call, the subnet addresses of
different IP sets are compared.

15   If the comparison shows that the subnet addresses
differ, then the default is set using G729A as the preferred
CODEC type for both IP end-point devices, although other CODECs
could be chosen as the default CODEC. Otherwise, if the
comparison shows that the subnet addresses are the same, then the

20   default is set using G711 as the preferred CODEC type for both IP
sets, although other CODECS could be chosen as the default CODEC.

In addition, software tools are used to measure network
performance parameters (e.g., jitter, available bandwidth, delay,
packet loss, latency, etc). In obtaining such measurements, a

25   ping tool, a network trace tool and/or a packet loss measurement
tool may be used to obtain the necessary information for
selection of the default settings.

The Ping tool measures latency and bandwidth of the
network between IP end-point devices and/or between end-point

30   devices and the PBX. It may do so by measuring the round trip
delay between an IP end-point device and a test point. Bandwidth
is measured by measuring the throughput per second of a
communication link. In cases where a voice path is directly
between two IP end-point devices and not through an intermediary

gateway, the test point requests the IP end-point devices to make the measurement on its behalf with respect to latency and bandwidth between the two IP end-point devices that are connected together. The test results become known to the terminal proxy

5   server, which makes the decisions on these parameters on behalf of the IP end-point devices.

If the latency is low and the bandwidth is high, Table 2 is used to select a small number (about 1-4 frames by example only) of voice frames per packet. If the latency is high and the

10   bandwidth is low, Table 1 is used to select a larger number (about 7-10 frames by example only) of voice frames per packet for optimum efficiency.

If the bandwidth is low, such as at 64 kbs, a CODEC with a high compression ratio should be used (as exemplified by

15   G723.1) and the latency will always increase as a consequence. In general, the less bandwidth, the worse the latency will likely be. On the other hand, if the bandwidth is high, the latency could also be high if the CODEC selection, packet size and jitter buffer size is chosen poorly. By recognizing these conditions,

20   however, appropriate changes may be made to these parameters to optimize latency and voice quality.

If there is spare bandwidth, then network efficiency becomes less of an issue. As a result, stuffing a small number, such as less than three, frames per packet produces less

25   delay/latency and yields better voice quality.

The Network trace tool is used to determine the number of Router Hops between IP end-point devices and/or between the end-point devices and the PBX. If the number of hops is small (1-2 for example) and the bandwidth is high, a small number of

30   frames per packet is used from Table 2 for G711. Otherwise, if the number of hops is large and the bandwidth is small, a large number of frames per packet is selected from Table 1 for G729A.

Fig. 2 illustrates a three-phase approach to optimize a VoIP connection, namely, initializing, network performance

monitoring and dynamically intervening or correcting. The
initializing phase has been discussed in detail with respect to
Fig. 1 and involves optimizing the configuration of the end point
devices. The end point is a frame of reference and can be
5   regarded as either the terminal or the gateway.

Network performance monitoring involves periodically
performing (e.g., every few minutes, or other time intervals), a
maintenance task that measures the network performance. The
maintenance task may involve the use of the Ping tool, the
10   Network trace tool and/or the data packet loss measurement tool.
If the network is found to have changed significantly as compared
to its condition at the time of initialization, the steps
mentioned for Initialization with respect to Fig. 1 may be
repeated to re-optimize the network. This could be done
15   automatically or at the request of an end-point device.

The default parameters for the end-point devices should
be changed where the measured performance parameters signify that
connection to the network is below a desired level of operation.
If the latency exceeds 200 milliseconds, for instance, changes
20   should be made to the default parameters. For instance, changing
the default parameters for the packet size and CODEC type will
help. If the bandwidth is below a threshold level of 128 kbs,
the default parameters for the preferred CODEC type needs to be
changed. If the packet loss is more than 5%, the default
25   parameters should be changed to a low number of voice samples per
frame.

The terminal proxy server and the end-point devices
make the necessary measurements with the software tools and
report the findings back to the terminal proxy server, which uses
30   conventional protocol to instruct the end-point devices to make
the appropriate changes in their default parameter settings.
Conventional programming is used to implement the instructions
from the terminal proxy server.

The next phase involves dynamic intervention or

correction when the end point devices are active on a session.
For example., during an active session, when two end-point devices
are in voice communication, corrective measures may need to be
taken to improve voice quality. Using intelligence in the IP
5   sets, an indication of the voice quality is reported to a
centralized terminal proxy server (see Fig. 3).

          If the voice quality at either end-point device has
degraded below a set threshold value, then the steps mentioned in
connection with Fig. 1 at the time of initialization may be
10   automatically implemented. Such degradation signifies that
conditions of the network may have changed significantly as
compared to the time of initialization As the voice channel is
re-optimizing, a slight interruption may occur in the voice path.
The voice paths may be muted by the end points while this
15   transition is taking place. This would make the transition
seamless from an end user perspective. Preferably, the user is
prompted to acknowledge that such re-optimization action is to be
taken, although such prompting is not required.

          At the time of initialization, an IP-PBX may be used
20   that could set certain parameters for each device. The
parameters are the preferred CODEC, the number of Voice Samples
per IP packet, and the jitter buffer size. For VoIP end-point
devices that are installed on the same subnet, the IP-PBX may set
the parameters the same as follows. The CODEC selection is G711,
25   the number of voice samples that would be sent with each packet
is less than five (and thus small, yielding approximately 50%
overhead in the TCP/IP frame), and the jitter buffer selected
would be very small or none.

          This three-phase approach will give the best audio
30   quality with the lowest amount of latency and will use the
minimum amount of digital signal processor (DSP) resources. This
approach, however, does not ensure a good connection, because it
only optimizes the available connection. This approach works well
if there is sufficient bandwidth on the local subnet and if the

packet loss is small (this is very probable in this scenario).

For devices on a different Subnet, the IP-PBX may perform a number of tests before defaulting the system with the above parameters. However, to save processing time and power, it

5   could default the system to the above parameters and then perform the tests to adjust the default settings as necessary based on results from the tests.

By pinging the end-points with data, it is possible to measure the throughput, latency and packet loss of the network

10   connection. It is likely that a G729A CODEC, a much larger packet size (e.g., 20 voice sample) and a sizable jitter buffer to allow for out of order packets would be selected.

By continually monitoring the network it is possible to dynamically change the default parameters associated with a VoIP

15   end point so that when a new call is started or received the network connection is optimized.

If a network is experiencing performance issues during a VoIP call, the IP-PBX could either automatically re-negotiate the CODEC connection and re-set the parameters for packet size

20   and/or jitter buffer. The option to perform such dynamic intervention during media transmission could also be provided to the user with a manual interface to start this activity (e.g., similar to the "scan" button on a cordless phone).

Since the bandwidth of the spectrum is much more

25   limited for wireless devices than for that of a wired LAN, the default parameters could be set based on whether the WLAN was a direct sequence spread spectrum (DSSS) (such as 11 mbs) or a frequency hopping (FH) spread spectrum(such as 2 mbs) or other type of spread spectrum. DSSS and FH spread spectrum are two

30   types of modulation techniques for use in wireless communication systems. Depending upon system requirements, each modulation has its set of advantages and disadvantages. In a messaging environment, frequency hopping is more attractive than direct sequence because it requires no power control at a portable

subscriber unit or end-point device. DS is more attractive for location finding applications (using time of arrival) or where spectral reconstruction can be used in interference cancellation.

As a consequence of following this three-phase

5   approach, voice quality becomes optimized for those IP networks that have not been provisioned for quality of service (QoS) and is further optimized if the network employs QoS measures. This is a self-learning and self-correcting solution to maximize voice and other media performance in an IP network.

10   While the foregoing description and drawings represent the preferred embodiments of the present invention, it will be understood that various changes and modifications may be made without departing from the spirit and scope of the present invention.